WHAT IS CLAIMED IS:

1. A method of operating an output buffer in a system processing streaming data comprising the steps of:

determining a time period in number of samples required or available to process a plurality of data samples;

loading a number of phantom samples into the output buffer equivalent in time to the time period required or available to process the data samples;

streaming the phantom samples from the output buffer for driving an external device generating a presentation; and

concurrent with said step of streaming the phantom samples, processing and loading the data samples into the output buffer behind the phantom samples.

2. The method of Claim 1 further comprising the step of calculating a fullness value for the output buffer using a number representing at least some of the phantom samples.

3. The method of Claim 1 wherein said step of determining a time period comprises the step of determining a sample advantage representing a difference in number of samples being output between a presentation time for a reference sample and time of availability of the reference sample.

4. The method of Claim 1 and further comprising the step of obtaining a sufficient time advantage prior to said step of determining a time period, the time advantage representing a time period between a presentation time of a reference sample and time of availability of the reference sample.

5. The method of Claim 1 wherein the data samples comprise audio samples and the phantom samples represent silence.

6. The method of Claim 1 wherein the data samples comprise video samples and the phantom samples represent dark frames.

7. The method of Claim 1 and further comprising the steps of:

receiving a presentation time stamp associated with the data samples; and

outputting a selected one of said data samples from the output behind the phantom samples at a time indicated by the presentation time stamp to achieve a perfect start.

8. An audio decoder comprising:

an input port for receiving a stream of audio data;

a data buffer for storing audio samples retrieved from said stream;

an output first-in-first-out memory for sourcing decoded audio data to an external device at a selected sampling rate and loaded from the data buffer when said first-in-first-out memory reaches a partially empty level; and

a digital signal processor operable to pre-fill said output memory by:

determining a sample advantage representing a difference in number of samples between a presentation time for a reference sample and time of availability of the reference sample;

loading a number of phantom samples into the output memory equivalent to the sample advantage;

streaming the phantom samples from the output memory at the sampling rate; and

during the streaming of the phantom samples, decompressing and loading into the output memory a plurality of data samples.

9. The decoder of Claim 8 wherein the digital signal processor is further operable to calculate a dipstick value monitoring the empty level of the output memory, at least some of the phantom samples contributing to the calculation.

10. The decoder of Claim 8 wherein the phantom samples represent silence samples.

11. The decoder of Claim 8 wherein the digital signal processor is further operable to selectively discard integrally encoded units of data to maximize the sample advantage and maximize pre-fill of the output memory.

12. The decoder of Claim 8 wherein the stream of audio data comprises a packetized elementary data stream.

13. The decoder of Claim 8 wherein said digital signal processor is a selected one of a plurality of digital signal processors forming a portion of said decoder.

14. The decoder of Claim 8 wherein said digital signal processor pre-fills said output memory at a start of a presentation.

15. The decoder of Claim 8 wherein said digital signal processor pre-fills said output memory after a change of channel.

16. The decoder of Claim 8 wherein said digital signal processor pre-fills said output memory following a loss of synchronization.

17. A method of processing a stream of encoded units of data samples comprising the steps of:

calculating a sample advantage using timing information embedded in selected ones of the encoded units, the sample advantage representing a difference in number of samples between the presentation of a reference sample and the availability of the reference sample for output;

queuing a number of phantom samples substantially equal to the number of samples represented by the calculated sample advantage;

outputting the phantom samples from the queue at a selected rate; and

decoding at least some data samples of at least one encoded unit and queuing the decoded data samples behind the phantom samples substantially simultaneously with said step of outputting.

18. The method of Claim 17 wherein said step of queuing comprises the step of queuing samples in a first-in-first-out memory.

19. The method of Claim 17 wherein a selected one of the decoded samples is output from the queue behind the phantom samples at a time indicated by the timing information to achieve a perfect start.

20. The method of Claim 17 further comprising the steps of:

calculating a value representing a number of samples in the queue using selected ones of the queued phantom samples; and

queuing selected ones of the data samples when the calculated value falls below a preselected threshold.